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PCT/DE00/02411INTERNATIONAL FILING DATE  
(20.07.00)  
20 July 2000PRIORITY DATE(S) CLAIMED  
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23 July 1999TITLE OF INVENTION  
**METHOD OF ADAPTIVE ADJUSTMENT OF THE COEFFICIENTS OF AN EQUALIZER**

APPLICANT(S) FOR DO/EO/US

**HERBIG, Gerhard; and GEBAUER, Thomas**

Applicant(s) herewith submit to the United States Designated/Elected Office (DO/EO/US) the following items and other information

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)) immediately rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39(1).
4. ☒ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.
5. ☒ A copy of the International Application as filed (35 U.S.C. 371(c)(2))
  - a. ☐ is transmitted herewith (required only if not transmitted by the International Bureau).
  - b. ☒ has been transmitted by the International Bureau.
  - c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US)
6. ☒ A translation of the International Application into English (35 U.S.C. 371(c)(2)).
7. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3))
  - a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau).
  - b. ☐ have been transmitted by the International Bureau.
  - c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
  - d. ☒ have not been made and will not be made.
8. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
9. ☒ An oath or declaration of the inventor(s) (35 U.S.C. 371(c)(4)) (unsigned).
10. ☒ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371(c)(5)).

**Items 11. to 16. below concern other document(s) or information included:**

11. ☒ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
12. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
13. ☒ A **FIRST** preliminary amendment.  
☐ A **SECOND** or **SUBSEQUENT** preliminary amendment.
14. ☒ A substitute specification and a marked up version thereof.
15. ☐ A change of power of attorney and/or address letter.
16. ☒ Other items or information: International Search Report, International Preliminary Examination Report and Form PCT/RO/101.



**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE**

Applicant(s) : Gerhard HERBIG et al.  
Serial No. : To Be Assigned  
Filed : Herewith  
For : METHOD OF ADAPTIVE ADJUSTMENT  
OF THE COEFFICIENTS OF AN EQUALIZER  
  
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Examiner : To Be Assigned

Assistant Commissioner  
for Patents  
Washington, D.C. 20231

**PRELIMINARY AMENDMENT AND  
37 C.F.R. § 1.125 SUBSTITUTE SPECIFICATION STATEMENT**

SIR:

Please amend without prejudice the above-identified application before examination, as set forth below.

**IN THE TITLE:**

Please amend without prejudice the title to be:  
--METHOD OF ADAPTIVE ADJUSTMENT OF THE COEFFICIENTS OF AN EQUALIZER--

**IN THE SPECIFICATION AND ABSTRACT:**

In accordance with 37 C.F.R. § 1.121(b)(3), a Substitute Specification (including the Abstract, but without claims) accompanies this response. It is respectfully requested that the Substitute Specification (including Abstract) be entered to replace the Specification of record.

**IN THE CLAIMS:**

Without prejudice, please cancel original claims 1 to 3 and new/substitute claims 1 to 3, and please add new claims 4 to 15 as follows:

--4. (New) A method of adaptively adjusting coefficients of an equalizer, the method comprising:

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defining the coefficients according to an error correction algorithm to minimize intersymbol interference; and

adjusting the coefficients, determined with the help of the error correction algorithm, by a correction term so that one of (i) a transfer function of the equalizer and (ii) at least one of a first derivation of the transfer function and a higher derivation of the transfer function assumes an invariant fixed value for at least two selected frequencies outside a useful signal frequency band.

5. (New) The method of claim 4, wherein the coefficients are adjusted so that the one of (i) the transfer function and (ii) the at least one of the first derivation of the transfer function and the higher derivation of the transfer function assumes a fixed value at at least two of the frequencies  $2\pi/T$ ,  $3\pi/T$  and  $4\pi/3T$ , and  $\pi/T$  is a limit frequency of the useful signal frequency band.

6. (New) The method of claim 4, wherein the one of (i) the transfer function and (ii) the first derivation of the transfer function and the higher derivation of the transfer function is set at one of a 0 value and at another constant value at the at least two selected frequencies.

7. (New) The method of claim 5, wherein the one of (i) the transfer function and (ii) the first derivation of the transfer function and the higher derivation of the transfer function is set at one of a 0 value and at another constant value at the at least two selected frequencies.

8. (New) An apparatus for adaptively adjusting coefficients of an equalizer, the apparatus comprising:

a first arrangement to define the coefficients according to an error correction algorithm to minimize intersymbol interference; and

a second arrangement to adjust the coefficients, determined with the help of the error correction algorithm, by a correction term so that one of (i) a transfer function of the equalizer and (ii) at least one of a first derivation of the transfer function and a higher derivation of the transfer function assumes an invariant fixed value for at least two selected frequencies outside a useful signal frequency band.

9. (New) The apparatus of claim 8, wherein the coefficients are adjusted so that the one of (i) the transfer function and (ii) the at least one of the first derivation of the transfer function and the

higher derivation of the transfer function assumes a fixed value at at least two of the frequencies  $2\pi/T$ ,  $3\pi/T$  and  $4\pi/3T$ , and  $\pi/T$  is a limit frequency of the useful signal frequency band.

10. (New) The apparatus of claim 8, wherein the one of (i) the transfer function and (ii) the first derivation of the transfer function and the higher derivation of the transfer function is set at one of a 0 value and at another constant value at the at least two selected frequencies.

11. (New) The apparatus of claim 9, wherein the one of (i) the transfer function and (ii) the first derivation of the transfer function and the higher derivation of the transfer function is set at one of a 0 value and at another constant value at the at least two selected frequencies.

12. (New) An apparatus for adaptively adjusting coefficients of an equalizer, the apparatus comprising:

means for defining the coefficients according to an error correction algorithm to minimize intersymbol interference; and

means for adjusting the coefficients, determined with the help of the error correction algorithm, by a correction term so that one of (i) a transfer function of the equalizer and (ii) at least one of a first derivation of the transfer function and a higher derivation of the transfer function assumes an invariant fixed value for at least two selected frequencies outside a useful signal frequency band.

13. (New) The apparatus of claim 12, wherein the coefficients are adjusted so that the one of (i) the transfer function and (ii) the at least one of the first derivation of the transfer function and the higher derivation of the transfer function assumes a fixed value at at least two of the frequencies  $2\pi/T$ ,  $3\pi/T$  and  $4\pi/3T$ , and  $\pi/T$  is a limit frequency of the useful signal frequency band.

14. (New) The apparatus of claim 12, wherein the one of (i) the transfer function and (ii) the first derivation of the transfer function and the higher derivation of the transfer function is set at one of a 0 value and at another constant value at the at least two selected frequencies.

15. (New) The apparatus of claim 13, wherein the one of (i) the transfer function and (ii) the first derivation of the transfer function and the higher derivation of the transfer function is set at one of a 0 value and at another constant value at the at least two selected frequencies.--.

### Remarks

This Preliminary Amendment cancels without prejudice original claims 1 to 3 and new/substitute claims 1 to 3 in the underlying PCT Application No. PCT/DE00/02411, and adds without prejudice new claims 4 to 15. The new claims conform the claims to U.S. Patent and Trademark Office rules and do not add new matter to the application.

In accordance with 37 C.F.R. § 1.121(b)(3), the Substitute Specification (including the Abstract, but without the claims) contains no new matter. The amendments reflected in the Substitute Specification (including Abstract) are to conform the Specification and Abstract to U.S. Patent and Trademark Office rules or to correct informalities. As required by 37 C.F.R. § 1.121(b)(3)(iii) and § 1.125(b)(2), a Marked Up Version Of The Substitute Specification comparing the Specification of record and the Substitute Specification also accompanies this Preliminary Amendment. In the Marked Up Version, underlining indicates added text and bracketing indicated deleted text. Approval and entry of the Substitute Specification (including Abstract) is respectfully requested.

The underlying PCT Application No. PCT/DE00/02411 includes an International Search Report, dated December 28, 2000. The Search Report includes a list of documents that were uncovered in the underlying PCT Application. A copy of the Search Report accompanies this Preliminary Amendment.

The underlying PCT application also includes an International Preliminary Examination Report, dated December 17, 2001, and an annex (including new/substitute claims 1 to 3). An English translation of the International Preliminary Examination Report and the annex accompanies this Preliminary Amendment.

Applicants assert that the subject matter of the present application is new, non-obvious, and useful. Prompt consideration and allowance of the application are respectfully requested.

Dated: 1/23/2002

Respectfully Submitted,  
KENYON & KENYON

By: Richard L. Mayer

Richard L. Mayer  
(Reg. No. 22,490)

One Broadway  
New York, NY 10004  
(212) 425-7200

**CUSTOMER NO. 26646**

(By) Richard L. Mayer  
Reg. No. 22,490  
33,865  
Jaron C  
DENISCH)

442511

[10191/2232]

METHOD OF ADAPTIVE ADJUSTMENT OF THE COEFFICIENTS  
OF AN EQUALIZER

[Background Information]

]FIELD OF THE INVENTION

The present invention relates to a method of adaptive adjustment of the coefficients of an equalizer, the coefficients being adapted according to an error correction algorithm so that intersymbol interference is minimized.

BACKGROUND INFORMATION

When digital signals are to be transmitted over channels having time-variant channel distortion, there [must] may be adaptive equalizers in the receiver, automatically adapting to the channel distortion to compensate it. Channel distortion may occur[s], for example, on radio transmission channels in point-to-point radio relay connections based on multiway propagation. A distinction [is] may be made between baud-spaced equalizers and fractionally spaced equalizers. In the case of baud-spaced equalizers, there [is] may be exactly one sample at both the input and the output of each symbol clock pulse. Since the sampling condition is already violated at the input of the equalizer in this case, such equalizers may have only limited capacity. [ M]

It is believed that much better results [are] may be obtained with a fractionally spaced equalizer, the simplest embodiment of which may proces[se]s exactly two samples per symbol clock pulse at the input. Even with a fractionally spaced equalizer, only ever one sample [is] may be calculated per symbol clock pulse at the output because there [is] may also be only ever one transmission symbol to be detected per symbol clock pulse.

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Since[ the usual] error correction algorithms for adaptive adjustment of the equalizer coefficients [can]may analyze only the baud-spaced output signal, fundamentally important information regarding the complete control of all possible coefficient settings [is]may be missing in the case of fractionally spaced equalizers. As a result, this type of equalizer may tend[s] toward unwanted "coefficient drift" which can no longer be controlled with the usual algorithms. [C]As to coefficient drift[ means that], because of rounding errors in the calculation of the coefficients, there [are]may be very gradual changes in the adapted correction quantities for the coefficient values in one direction. [In general, t]The following [known]available algorithms [are]may be used for adaptation of the equalizer coefficients of fractionally spaced equalizers.

During the acquisition phase, as long as the transmission carrier has not yet been recognized, so the closed loop is not yet engaged for carrier phase synchronization, the constant modulus algorithm (CMA) [is usually]may be used, and the [least]"least means square" algorithm (LMSA) [is]may be used during the tracking phase, i.e., in the actual continuous operation of the receiver. The two aforementioned algorithms are [described]referred to, for example, in K. D. Kammeyer, Nachrichtenübertragung [Telecommunication], B. G. Teubner Verlag, Stuttgart, 1992, pp. 313-316, 510-512 and in J. G. Proakis, Digital Communications, McGraw-Hill, 1989, pp. 561-569, 587-593.

Coefficient drift [is]may be observed with the CMA algorithm in particular as a result of quantizing errors, because it [is]may be difficult especially with CMA, which is a higher order algorithm, to completely prevent offset errors as a result of quantization operations. With the LMSA algorithm, the lack of control capability, especially with dynamic

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transmission channels, may result[s] in inadequate adaptation of the equalizer coefficients to the channel changes. Especially channels with multiway reception, such as that with point-to-point radio relay connections, may result in failure of the equalizer in cases of less serious distortion, although in terms of its basic capability, it [would]should be quite capable of equalizing these channels.

The tap leakage algorithm (TLA), which is a variant of LMSA, is a countermeasure against the phenomenon of coefficient drift in a fractionally spaced equalizer. The tap leakage algorithm is [described by]referred to in R. D. Giltin, H. C. Meadors, S. B. Weinstein: The Tap Leakage Algorithm: an Algorithm for the Stable Operation of a Digitally Implemented, Fractionally Spaced Adaptive Equalizer. BSDJ, no. 8, vol. 61, October 1982, pages 1817 through 1839. According to the TLA, smaller amounts are subtracted from the absolute value of the coefficients to compensate for coefficient drift. However, this measure may not only prevent[s] coefficient drift but may also ha[s]ve a negative effect on the equalization result, i.e., there [is]may be an increase in intersymbol interference.

[Therefore, the object]SUMMARY OF THE INVENTION

An object of an exemplary embodiment and/or exemplary method of the present invention is to provide a method[ of the type defined in the preamble] with which coefficient drift may be prevented without having or at least limiting a negative effect on the quality of equalization.

[Advantages of the Invention

This object is achieved with the features of Claim 1 by the fact that]In this regard, the coefficients determined with the help of the error correction algorithm are adjusted by a

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correction term so that the transfer function of the equalizer assumes an invariant fixed value for one or more selected frequencies outside the useful signal frequency band. Instead of the transfer function itself, a first derivation and/or a higher derivation of the transfer function outside the useful signal frequency band may be set at a fixed value for one or more selected frequencies. [T]It is believed that this method should largely [reduces]reduce or at least reduce overshooting of the transfer function of the equalizer toward both sides of the useful signal frequency band which [can]may be attributed to unwanted coefficient drift. This may prevent[s] dynamic fading events on the transmission link from leading to failure of the equalizer, e.g., when a certain coefficient setting can no longer be adapted rapidly enough.

[Advantageous refinements of the present invention are derived from the subclaims. The simplest]It is believed that a simpler, if not the simplest, possible method of calculating correction terms for the coefficients [is possible]may be obtained if the transfer function or the first derivation and/or a higher derivation of the transfer function at the frequency  $2\pi/T$  and/or the frequency  $3\pi/T$  and/or the frequency  $4\pi/3T$  is set at a fixed value, where  $\pi/T$  is the base frequency of the useful signal frequency band. The transfer function or a first derivation and/or a higher derivation thereof may be set at the value 0 or at another fixed value at the selected frequency or frequencies.

[Drawing

The present invention is explained in greater detail with reference to an exemplary embodiment.

Figure 1 ]BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 shows a block diagram of an adaptive equalizer[;].

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Figure 2 [ ] shows a transfer function of the equalizer without the correction according to an exemplary embodiment and/or exemplary method of the present invention[;]\_.

Figure 3 [ ] shows a transfer function of the equalizer having a first correction[, and]\_.

Figure 4 [ ] shows a transfer function of the equalizer having a second correction.

[Description of a Embodiment

] DETAILED DESCRIPTION

Figure 1 shows an adaptive transversal equalizer having a time-delay chain, of which first two time-delay elements  $V_0$  and  $V_1$  are shown here. A digital input signal  $x$ , which has been distorted on the transmission link from a transmitter to the receiver in which the adaptive equalizer is located, is applied to input 1 of the time-delay chain. Input signal  $x$  is delayed by  $T/2$ ,  $T$  being the symbol clock pulse of input signal  $x$ , in individual time-delay elements  $V_0$ ,  $V_1$ . Before the individual time-delay elements, the symbols of distorted input signal  $x$  are picked up from the time-delay chain and sent to a multiplier  $M_0$ ,  $M_1$ , where the symbol is weighted with a coefficient  $w(0)$ ,  $w(1)$ . [ ]

All signal symbols  $y(0)$ ,  $y(1)$ , ...  $y(n)$  formed in the equalizer and weighted with coefficients  $w(0)$ ,  $w(1)$ , ...  $w(n)$  in this way are combined to an output signal  $y$  by a summing unit. Output signal  $y$  is sent to a decision circuit ES which decides for each symbol of output signal  $y$  which of the possible transmission symbols it most closely approximates, i.e., decision circuit ES estimates the transmission symbols most likely to be sent on the basis of the symbols of summing

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unit output signal  $y$ . Estimated transmission symbols  $[a]$  "a" may be picked up at output 2 of decision circuit ES. Decision circuit ES also generates an error signal  $e$  which depends on the deviation between the respective symbol of summing unit output signal  $y$  and estimated transmission symbol  $a$ .

Error signal  $e$  is sent to correlators  $K0$ ,  $K1$  which [are responsible for forming] form coefficients  $w(0)$ ,  $w(1)$ . Correlators  $K0$ ,  $K1$  determine adaptive change values for coefficients  $w(0)$ ,  $w(1)$  according to a known error correction algorithm, e.g., according to the LMSA algorithm mentioned above, from error signal  $e$  and the symbols of distorted input signal  $x$  picked up from the time-delay chain. Each correlator  $K0$ ,  $K1$  is followed by an adder  $A0$ ,  $A1$  in which a correction term  $kt(0)$ ,  $kt(1)$  is added to the change value output by correlator  $K0$ ,  $K1$  for the coefficients  $w(0)$ ,  $w(1)$ . Correction terms  $kt(0)$ ,  $kt(1)$  are formed in a processor (PZ) according to an algorithm [to be] further described[ in greater detail] below. These correction terms  $kt(0)$ ,  $kt(1)$  are aimed at preventing the coefficient drift mentioned above.

Individual adders  $A0$ ,  $A1$  are each followed by a coefficient register  $KR0$ ,  $KR1$  in which integration is performed over all change values including the correction terms for coefficients  $w(0)$ ,  $w(1)$ , which then yields the respective instantaneous coefficient  $w(0)$ ,  $w(1)$ .

Figure 1 shows an adaptive equalizer for an actual digital input signal  $x$ . However, QAM signals [are usually] may be emitted in a digital radio relay system. Accordingly, four such adaptive equalizers [would] may have to be provided for one QAM receiver, namely one in the in-phase branch, one in the quadrature-phase branch, and for compensation of crosstalk, one adaptive equalizer connected from the in-phase branch to the quadrature-phase branch and one connected from

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the quadrature-phase branch to the in-phase branch.

The following explains how processor PZ generates correction terms  $kt(k)$ , where  $k = 0, 1, \dots, n$ . As already indicated, coefficient drift is to be suppressed by correction terms  $kt(k)$  for coefficients  $w(k)$ . Figure 2 shows a transfer function  $E(\omega)$  of an equalizer, where the useful signal frequency band has its base frequencies at  $\omega = \pm\pi/T$ , but the equalizer may influence the spectrum in the range of  $\omega = -2\pi/T$  to  $\omega = +2\pi/T$ . Unwanted coefficient drift is manifested by a gain in the spectral ranges not used by the useful signal (shown in gray in Figure 2). Correction term  $kt(k)$  directly influences the transfer function of the equalizer, so that the overshooting outside the useful signal frequency band is reduced and therefore no more coefficient drift occurs. At the same time, however, it is believed that the transfer function within the useful signal frequency band remains completely (or at least substantially) unaffected by this, so that the actual equalization is not impaired, or should not be impaired as a practical matter.

The following equation holds for transfer function  $E(\omega)$  of the equalizer:

$$\begin{aligned} E(\omega) &= \sum (w_i(k) + jw_q(k)) e^{-j\omega k T/2} \\ &= E_i(\omega) + jE_q(\omega) \\ &= \sum w_i(k) \cos \omega k \frac{T}{2} + \sum w_q(k) \sin \omega k \frac{T}{2} \\ &\quad + j \left( -\sum w_i(k) \sin \omega k \frac{T}{2} + \sum w_q(k) \cos \omega k \frac{T}{2} \right) \end{aligned}$$

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]

(1)

[

]where  $w_i(k)$  and  $w_q(k)$  are the  $k$ -th in-phase coefficient and quadrature-phase coefficient, respectively.

5

Summation  $\Sigma$  extends over all coefficients from  $k = 0$  through  $k = n$ . The algorithm being run in processor PZ should form a zero place or another fixed value of the transfer function at at least one specific frequency  $\omega_0 \approx \pm\pi/T$  to reduce the transfer function outside the useful [ ]signal frequency band. This yields as the target function of the algorithm:

10

$$|E(\omega_0)| = |E_i(\omega_0)|^2 + |E_q(\omega_0)|^2 \quad (2)$$

15

With a complex equalizer for QAM signals, all the respective four partial equalizers are to be adjusted in their transfer function independently of one another in the sense described above, so the differentiation between  $w_i$  and  $w_q$  for the in-phase branch and the quadrature-phase branch may be omitted.

20

This yields the following for the absolute value of the transfer function:

$$|E(\omega_0)|^2 = \left| \sum w(k) \cos \omega_0 k \frac{T}{2} \right|^2 + \left| \sum w(k) \sin \omega_0 k \frac{T}{2} \right|^2 \quad (3)$$

[

25

]and for its gradient:

$$\frac{\partial}{\partial w(k)} |E(\omega_0)|^2 = 2|E_i(\omega_0)| \frac{\partial |E_i(\omega_0)|}{\partial w(k)} + 2|E_q(\omega_0)| \frac{\partial |E_q(\omega_0)|}{\partial w(k)} \quad (4)$$

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$$= 2\cos\omega_0 k \frac{T}{2} \sum w(k) \cos\omega_0 k \frac{T}{2} + 2\sin\omega_0 k \frac{T}{2} \sum w(k) \sin\omega_0 k \frac{T}{2}$$

The algorithm for correction of coefficients  $w(k)$  is then formed as follows:

$$w_{n+1}(k) = w_n(k) - \alpha \cdot \text{sign}[J(k)] \quad (5)$$

$w_n(k)$  is the coefficient formed one clock pulse previously, and  $w_{n+1}(k)$  is the instantaneous coefficient derived by addition of correction term  $kt(k) = -\alpha \cdot \text{sign}[J(k)]$  to coefficient  $w_n(k)$ .

Otherwise, the change value for coefficient  $w_n(k)$  formed according to the [known] CMA or LMSA algorithms, for example, in correlators K0, K1 is not taken into account in equation (5) for the sake of simplicity.

In equation (5):

$$J(k) = \frac{\partial}{\partial w(k)} |E(\omega_0)|^2 \quad (6)$$

$$J(k) = W_C \cdot \cos\omega_0 k \frac{T}{2} + W_S \cdot \sin\omega_0 k \frac{T}{2}$$

where the abbreviations

$$\begin{aligned} W_C &= \sum w(k) \cos\omega_0 k \frac{T}{2} \\ W_S &= \sum w(k) \sin\omega_0 k \frac{T}{2} \end{aligned} \quad (7)$$

where were used. Since only a slight influence on the coefficients

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is desired (small  $\alpha$ ), the algorithm may be simplified in practical operation to a signum form. Effectiveness factor  $\alpha$  for correction term  $kt(k)$  is adjusted to a suitable value by field simulation.

5

[T] Since the algorithm [is to] should be implementable in a relatively simpler manner without any [great] "great effort[. Therefore,]", it is believed that it must be limited to frequencies  $\omega_0$  which permit a simple and most periodic possible calculation of the trigonometric functions. The following frequencies may be used for this:

10

$$\omega_A = \frac{2\pi}{T}, \omega_B = \frac{3\pi}{2T} \text{ and } \omega_C = \frac{4\pi}{3T}. \quad (8)$$

15

Two simplest cases  $\omega_A$  and  $\omega_C$  shall now be discussed.

1<sup>st</sup> case:  $\omega_A = 2 \pi/T$

For this especially simple case, it holds that

20

$$\begin{aligned} \sin \omega_A k \frac{T}{2} &= \sin k\pi = 0 \\ \cos \omega_A k \frac{T}{2} &= \cos k\pi = (-1)^k \end{aligned} \quad (10)$$

and therefore

$$\frac{\partial}{\partial w(k)} |E(\omega_A)|^2 = 2(-1)^k \cdot \sum w(k)(-1)^k \quad (10)$$

25

The complete algorithm for adaptation of the coefficients is

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thus written as follows according to equation (5):

$$w_{n+1}(k) = w_n(k) - 2\alpha \cdot \text{sign} \left[ (-1)^k \sum w(k)(-1)^k \right] \quad (11)$$

5 where

$$J_A = 2(-1)^k \cdot \sum w_n(k)(-1)^k \quad (12)$$

10 Since the middle coefficients [are preferably] may be involved in the unwanted components of the transfer function, the algorithm according to equation (11) may be limited to an interval on the order of  $k \in [-4, 4]$ . All coefficients must of course be taken into account in equation (12).

15 Figure 3 shows the result of this algorithm, where it is apparent that there has already been a certain reduction in the transfer function outside the useful signal frequency band in comparison with the uncorrected transfer function shown in Figure 1.

20 [

$$] 2^{\text{nd}} \text{ case: } \omega_c = \frac{4\pi}{3T}$$

[

] For this case, it holds that

$$\begin{aligned} \sin \omega_c k \frac{T}{2} &= \sin \frac{2}{3} k \pi = \Im(z^k) \\ \cos \omega_c k \frac{T}{2} &= \cos \frac{2}{3} k \pi = \Re(z^k) \end{aligned} \quad (13)$$

$$z = e^{j2\pi/3} = e^{j120^\circ} \quad [$$

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]With the table:

[

]

K	-4	-3	-2	-1	0	1	2	3	4
$\sin \frac{2}{3} k \pi$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$
$\cos \frac{2}{3} k \pi$	-0,5	1	-0,5	-0,5	1	-0,5	-0,5	1	-0,5

[

]it holds that:

$$W_C = \sum w(k) \cos \omega_c k \frac{T}{2} = \sum w(3k) - 0,5 \sum [w(3k+1) + w(3k-1)]$$

] (14)

$$W_S = \sum w(k) \sin \omega_c k \frac{T}{2} = \sqrt{0,75} \sum [w(3k+1) - w(3k-1)]$$

] (15)

[

]To calculate the correction terms, additional abbreviations are introduced:

$$W_{-1} = \sum w(3k-1)$$

$$W_0 = \sum w(3k)$$

$$W_{+1} = \sum w(3k+1)$$

(16)

$$WD = W_{+1} - W_{-1}$$

$$WS = W_{+1} + W_{-1}$$

[

**MARKED UP VERSION OF SUBSTITUTE SPECIFICATION**

] It thus holds that:

$$J_C(3k-1) = -\sqrt{0,75} \cdot W_S - 0,5 \cdot W_C = \frac{1}{4}(-3WD - 2W_0 + WS)$$

$$J_C(3k) = W_C \quad [$$

$$J_C(3k+1) = +\sqrt{0,75} \cdot W_S - 0,5 \cdot W_C = \frac{1}{4}(+3WD - 2W_0 + WS)$$

] (17)

Whereas in the first case,  $J_A$  according to equation (12) holds for all coefficients  $w(k)$ , in the second case  $J_C$  must be different for three different coefficient groups  $w(3k-1)$ ,  $w(3k)$  and  $w(3k+1)$ .

A very efficient correction of the coefficients is obtained when the first and second cases are combined:

$$w_{n+1}(k) = w_n(k) - \alpha_A \cdot \text{sign}[J_A(k)] - \alpha_C \cdot \text{sign}[J_C(k)] \quad (18)$$

As Figure 4 shows, the adaptive correction of coefficients  $w_n(k)$  according to equation (18) causes a sharp reduction in the spectral ranges outside the useful signal spectrum.

Instead of forcing zero places in the transfer function through the correction, as shown in Figures 3 and 4, the transfer function may also be set at a constant value (e.g. 1) at certain frequencies.

Instead of setting the transfer function of the equalizer itself at a constant value at certain frequencies, a first derivation and/or a higher derivation of the transfer function may also be set at a constant value for one or more selected frequencies.

**MARKED UP VERSION OF SUBSTITUTE SPECIFICATION**

[Abstract]

ABSTRACT OF THE DISCLOSURE

The coefficients  $[(w(0), w(1))]$  of an equalizer are adapted according to an error correction algorithm so that intersymbol interference is minimized. To prevent coefficient drift in adaptive equalization, the coefficients  $[(w(0), w(1))]$  determined with the help of the error correction algorithm are adjusted by a correction term  $[(k(0), k(1))]$  so that the transfer function of the equalizer assumes a favorable value for one or more selected frequencies outside the useful signal frequency band.

[(Figure 1)]

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[10191/2232]

METHOD OF ADAPTIVE ADJUSTMENT OF THE COEFFICIENTS  
OF AN EQUALIZERFIELD OF THE INVENTION

The present invention relates to a method of adaptive adjustment of the coefficients of an equalizer, the coefficients being adapted according to an error correction algorithm so that intersymbol interference is minimized.

BACKGROUND INFORMATION

When digital signals are to be transmitted over channels having time-variant channel distortion, there may be adaptive equalizers in the receiver, automatically adapting to the channel distortion to compensate it. Channel distortion may occur, for example, on radio transmission channels in point-to-point radio relay connections based on multiway propagation. A distinction may be made between baud-spaced equalizers and fractionally spaced equalizers. In the case of baud-spaced equalizers, there may be exactly one sample at both the input and the output of each symbol clock pulse. Since the sampling condition is already violated at the input of the equalizer in this case, such equalizers may have only limited capacity.

It is believed that much better results may be obtained with a fractionally spaced equalizer, the simplest embodiment of which may process exactly two samples per symbol clock pulse at the input. Even with a fractionally spaced equalizer, only ever one sample may be calculated per symbol clock pulse at the output because there may also be only ever one transmission symbol to be detected per symbol clock pulse.

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Since error correction algorithms for adaptive adjustment of the equalizer coefficients may analyze only the baud-spaced output signal, fundamentally important information regarding the complete control of all possible coefficient settings may be missing in the case of fractionally spaced equalizers. As a result, this type of equalizer may tend toward unwanted "coefficient drift" which can no longer be controlled with the usual algorithms. As to coefficient drift, because of rounding errors in the calculation of the coefficients, there may be very gradual changes in the adapted correction quantities for the coefficient values in one direction. The following available algorithms may be used for adaptation of the equalizer coefficients of fractionally spaced equalizers.

During the acquisition phase, as long as the transmission carrier has not yet been recognized, so the closed loop is not yet engaged for carrier phase synchronization, the constant modulus algorithm (CMA) may be used, and the "least means square" algorithm (LMSA) may be used during the tracking phase, i.e., in the actual continuous operation of the receiver. The two aforementioned algorithms are referred to, for example, in K. D. Kammeyer, Nachrichtenübertragung [Telecommunication], B. G. Teubner Verlag, Stuttgart, 1992, pp. 313-316, 510-512 and in J. G. Proakis, Digital Communications, McGraw-Hill, 1989, pp. 561-569, 587-593.

Coefficient drift may be observed with the CMA algorithm in particular as a result of quantizing errors, because it may be difficult especially with CMA, which is a higher order algorithm, to completely prevent offset errors as a result of quantization operations. With the LMSA algorithm, the lack of control capability, especially with dynamic transmission channels, may result in inadequate adaptation of the equalizer coefficients to the channel changes. Especially channels with multiway reception, such as that with point-to-point radio relay connections, may result in failure of the equalizer in

cases of less serious distortion, although in terms of its basic capability, it should be quite capable of equalizing these channels.

5 The tap leakage algorithm (TLA), which is a variant of LMSA, is a countermeasure against the phenomenon of coefficient drift in a fractionally spaced equalizer. The tap leakage algorithm is referred to in R. D. Giltin, H. C. Meadors, S. B. Weinstein: The Tap Leakage Algorithm: an Algorithm for the Stable Operation of a Digitally Implemented, Fractionally Spaced Adaptive Equalizer. BSDJ, no. 8, vol. 61, October 1982, pages 1817 through 1839. According to the TLA, smaller amounts are subtracted from the absolute value of the coefficients to compensate for coefficient drift. However, this measure may not only prevent coefficient drift but may also have a negative effect on the equalization result, i.e., there may be an increase in intersymbol interference.

#### SUMMARY OF THE INVENTION

20 An object of an exemplary embodiment and/or exemplary method of the present invention is to provide a method with which coefficient drift may be prevented without having or at least limiting a negative effect on the quality of equalization.

25 In this regard, the coefficients determined with the help of the error correction algorithm are adjusted by a correction term so that the transfer function of the equalizer assumes an invariant fixed value for one or more selected frequencies outside the useful signal frequency band. Instead of the transfer function itself, a first derivation and/or a higher derivation of the transfer function outside the useful signal frequency band may be set at a fixed value for one or more selected frequencies. It is believed that this method should largely reduce or at least reduce overshooting of the transfer function of the equalizer toward both sides of the useful signal frequency band which may be attributed to unwanted

#### SUBSTITUTE SPECIFICATION

coefficient drift. This may prevent dynamic fading events on the transmission link from leading to failure of the equalizer, e.g., when a certain coefficient setting can no longer be adapted rapidly enough.

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It is believed that a simpler, if not the simplest, possible method of calculating correction terms for the coefficients may be obtained if the transfer function or the first derivation and/or a higher derivation of the transfer function at the frequency  $2\pi/T$  and/or the frequency  $3\pi/T$  and/or the frequency  $4\pi/3T$  is set at a fixed value, where  $\pi/T$  is the base frequency of the useful signal frequency band. The transfer function or a first derivation and/or a higher derivation thereof may be set at the value 0 or at another fixed value at the selected frequency or frequencies.

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#### BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 shows a block diagram of an adaptive equalizer.

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Figure 2 shows a transfer function of the equalizer without the correction according to an exemplary embodiment and/or exemplary method of the present invention.

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Figure 3 shows a transfer function of the equalizer having a first correction.

Figure 4 shows a transfer function of the equalizer having a second correction.

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#### DETAILED DESCRIPTION

Figure 1 shows an adaptive transversal equalizer having a time-delay chain, of which first two time-delay elements  $V_0$  and  $V_1$  are shown here. A digital input signal  $x$ , which has been distorted on the transmission link from a transmitter to the receiver in which the adaptive equalizer is located, is applied to input 1 of the time-delay chain. Input signal  $x$  is

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delayed by  $T/2$ ,  $T$  being the symbol clock pulse of input signal  $x$ , in individual time-delay elements  $V_0, V_1$ . Before the individual time-delay elements, the symbols of distorted input signal  $x$  are picked up from the time-delay chain and sent to a multiplier  $M_0, M_1$ , where the symbol is weighted with a coefficient  $w(0), w(1)$ .

All signal symbols  $y(0), y(1), \dots y(n)$  formed in the equalizer and weighted with coefficients  $w(0), w(1), \dots w(n)$  in this way are combined to an output signal  $y$  by a summing unit. Output signal  $y$  is sent to a decision circuit ES which decides for each symbol of output signal  $y$  which of the possible transmission symbols it most closely approximates, i.e., decision circuit ES estimates the transmission symbols most likely to be sent on the basis of the symbols of summing unit output signal  $y$ . Estimated transmission symbols "a" may be picked up at output 2 of decision circuit ES. Decision circuit ES also generates an error signal  $e$  which depends on the deviation between the respective symbol of summing unit output signal  $y$  and estimated transmission symbol  $a$ .

Error signal  $e$  is sent to correlators  $K_0, K_1$  which form coefficients  $w(0), w(1)$ . Correlators  $K_0, K_1$  determine adaptive change values for coefficients  $w(0), w(1)$  according to a known error correction algorithm, e.g., according to the LMSA algorithm mentioned above, from error signal  $e$  and the symbols of distorted input signal  $x$  picked up from the time-delay chain. Each correlator  $K_0, K_1$  is followed by an adder  $A_0, A_1$  in which a correction term  $kt(0), kt(1)$  is added to the change value output by correlator  $K_0, K_1$  for the coefficients  $w(0), w(1)$ . Correction terms  $kt(0), kt(1)$  are formed in a processor (PZ) according to an algorithm further described below. These correction terms  $kt(0), kt(1)$  are aimed at preventing the coefficient drift mentioned above.

Individual adders  $A_0, A_1$  are each followed by a coefficient

register KR0, KR1 in which integration is performed over all change values including the correction terms for coefficients  $w(0)$ ,  $w(1)$ , which then yields the respective instantaneous coefficient  $w(0)$ ,  $w(1)$ .

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Figure 1 shows an adaptive equalizer for an actual digital input signal  $x$ . However, QAM signals may be emitted in a digital radio relay system. Accordingly, four such adaptive equalizers may have to be provided for one QAM receiver, namely one in the in-phase branch, one in the quadrature-phase branch, and for compensation of crosstalk, one adaptive equalizer connected from the in-phase branch to the quadrature-phase branch and one connected from the quadrature-phase branch to the in-phase branch.

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The following explains how processor PZ generates correction terms  $kt(k)$ , where  $k = 0, 1, \dots, n$ . As already indicated, coefficient drift is to be suppressed by correction terms  $kt(k)$  for coefficients  $w(k)$ . Figure 2 shows a transfer function  $E(\omega)$  of an equalizer, where the useful signal frequency band has its base frequencies at  $\omega = \pm\pi/T$ , but the equalizer may influence the spectrum in the range of  $\omega = -2\pi/T$  to  $\omega = +2\pi/T$ . Unwanted coefficient drift is manifested by a gain in the spectral ranges not used by the useful signal (shown in gray in Figure 2). Correction term  $kt(k)$  directly influences the transfer function of the equalizer, so that the overshooting outside the useful signal frequency band is reduced and therefore no more coefficient drift occurs. At the same time, however, it is believed that the transfer function within the useful signal frequency band remains completely (or at least substantially) unaffected by this, so that the actual equalization is not impaired, or should not be impaired as a practical matter.

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The following equation holds for transfer function  $E(\omega)$  of the equalizer:

$$\begin{aligned}
E(\omega) &= \sum (w_i(k) + jw_q(k)) e^{-j\omega k T/2} \\
&= E_i(\omega) + jE_q(\omega) \\
&= \sum w_i(k) \cos \omega k \frac{T}{2} + \sum w_q(k) \sin \omega k \frac{T}{2} \\
&\quad + j \left( -\sum w_i(k) \sin \omega k \frac{T}{2} + \sum w_q(k) \cos \omega k \frac{T}{2} \right)
\end{aligned} \tag{1}$$

where  $w_i(k)$  and  $w_q(k)$  are the  $k$ -th in-phase coefficient and quadrature-phase coefficient, respectively.

Summation  $\Sigma$  extends over all coefficients from  $k = 0$  through  $k = n$ . The algorithm being run in processor PZ should form a zero place or another fixed value of the transfer function at at least one specific frequency  $\omega_0 \approx \pm\pi/T$  to reduce the transfer function outside the useful signal frequency band. This yields as the target function of the algorithm:

$$|E(\omega_0)| = |E_i(\omega_0)|^2 + |E_q(\omega_0)|^2 \tag{2}$$

With a complex equalizer for QAM signals, all the respective four partial equalizers are to be adjusted in their transfer function independently of one another in the sense described above, so the differentiation between  $w_i$  and  $w_q$  for the in-phase branch and the quadrature-phase branch may be omitted. This yields the following for the absolute value of the transfer function:

$$|E(\omega_0)|^2 = \left| \sum w(k) \cos \omega_0 k \frac{T}{2} \right|^2 + \left| \sum w(k) \sin \omega_0 k \frac{T}{2} \right|^2 \tag{3}$$

and for its gradient:

$$\frac{\partial}{\partial w(k)} |E(\omega_0)|^2 = 2|E_i(\omega_0)| \frac{\partial |E_i(\omega_0)|}{\partial w(k)} + 2|E_q(\omega_0)| \frac{\partial |E_q(\omega_0)|}{\partial w(k)} \quad (4)$$

$$= 2\cos\omega_0 k \frac{T}{2} \sum w(k) \cos\omega_0 k \frac{T}{2} + 2\sin\omega_0 k \frac{T}{2} \sum w(k) \sin\omega_0 k \frac{T}{2}$$

The algorithm for correction of coefficients  $w(k)$  is then formed as follows:

$$w_{n+1}(k) = w_n(k) - \alpha \cdot \text{sign}[J(k)] \quad (5)$$

$w_n(k)$  is the coefficient formed one clock pulse previously, and  $w_{n+1}(k)$  is the instantaneous coefficient derived by addition of correction term  $kt(k) = -a \cdot \text{sign}[J(k)]$  to coefficient  $w_n(k)$ . Otherwise, the change value for coefficient  $w_n(k)$  formed according to the CMA or LMSA algorithms, for example, in correlators K0, K1 is not taken into account in equation (5) for the sake of simplicity.

In equation (5):

$$J(k) = \frac{\partial}{\partial w(k)} |E(\omega_0)|^2 \quad (6)$$

$$J(k) = W_C \cdot \cos\omega_0 k \frac{T}{2} + W_S \cdot \sin\omega_0 k \frac{T}{2}$$

where the abbreviations

$$\begin{aligned} W_C &= \sum w(k) \cos\omega_0 k \frac{T}{2} \\ W_S &= \sum w(k) \sin\omega_0 k \frac{T}{2} \end{aligned} \quad (7)$$

were used. Since only a slight influence on the coefficients

is desired (small  $\alpha$ ), the algorithm may be simplified in practical operation to a signum form. Effectiveness factor  $\alpha$  for correction term  $kt(k)$  is adjusted to a suitable value by field simulation.

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Since the algorithm should be implementable in a relatively simpler manner without any "great effort", it is believed that it must be limited to frequencies  $\omega_0$  which permit a simple and most periodic possible calculation of the trigonometric functions. The following frequencies may be used for this:

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$$\omega_A = \frac{2\pi}{T}, \omega_B = \frac{3\pi}{2T} \text{ and } \omega_C = \frac{4\pi}{3T}. \quad (8)$$

Two simplest cases  $\omega_A$  and  $\omega_C$  shall now be discussed.

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1<sup>st</sup> case:  $\omega_A = 2\pi/T$

For this especially simple case, it holds that

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$$\begin{aligned} \sin \omega_A k \frac{T}{2} &= \sin k\pi = 0 \\ \cos \omega_A k \frac{T}{2} &= \cos k\pi = (-1)^k \end{aligned} \quad (10)$$

and therefore

$$\frac{\partial}{\partial w(k)} |E(\omega_A)|^2 = 2(-1)^k \cdot \sum w(k)(-1)^k \quad (10)$$

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The complete algorithm for adaptation of the coefficients is thus written as follows according to equation (5):

$$w_{n+1}(k) = w_n(k) - 2\alpha \cdot \text{sign}\left[(-1)^k \sum w(k)(-1)^k\right] \quad (11)$$

where

$$J_A = 2(-1)^k \cdot \sum w_n(k)(-1)^k \quad (12)$$

Since the middle coefficients may be involved in the unwanted components of the transfer function, the algorithm according to equation (11) may be limited to an interval on the order of  $k \in [-4, 4]$ . All coefficients must of course be taken into account in equation (12).

Figure 3 shows the result of this algorithm, where it is apparent that there has already been a certain reduction in the transfer function outside the useful signal frequency band in comparison with the uncorrected transfer function shown in Figure 1.

$$2^{\text{nd}} \text{ case: } \omega_c = \frac{4\pi}{3T}$$

For this case, it holds that

$$\begin{aligned} \sin \omega_c k \frac{T}{2} &= \sin \frac{2}{3} k \pi = \Im(z^k) \\ \cos \omega_c k \frac{T}{2} &= \cos \frac{2}{3} k \pi = \Re(z^k) \end{aligned} \quad (13)$$

$$z = e^{j2\pi/3} = e^{j120^\circ}$$

With the table:

K	-4	-3	-2	-1	0	1	2	3	4
$\text{Sin} \frac{2}{3} k\pi$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$
$\text{Cos} \frac{2}{3} k\pi$	-0,5	1	-0,5	-0,5	1	-0,5	-0,5	1	-0,5

it holds that:

$$W_C = \sum w(k) \cos \omega_c k \frac{T}{2} = \sum w(3k) - 0,5 \sum [w(3k+1) + w(3k-1)] \quad (14)$$

$$W_S = \sum w(k) \sin \omega_c k \frac{T}{2} = \sqrt{0,75} \sum [w(3k+1) - w(3k-1)] \quad (15)$$

To calculate the correction terms, additional abbreviations are introduced:

$$\begin{aligned} W_{-1} &= \sum w(3k-1) \\ W_0 &= \sum w(3k) \\ W_{+1} &= \sum w(3k+1) \\ WD &= W_{+1} - W_{-1} \\ WS &= W_{+1} + W_{-1} \end{aligned} \quad (16)$$

It thus holds that:

$$\begin{aligned} J_C(3k-1) &= -\sqrt{0,75} \cdot W_S - 0,5 \cdot W_C = \frac{1}{4} (-3WD - 2W_0 + WS) \\ J_C(3k) &= W_C \\ J_C(3k+1) &= +\sqrt{0,75} \cdot W_S - 0,5 \cdot W_C = \frac{1}{4} (+3WD - 2W_0 + WS) \end{aligned} \quad (17)$$

Whereas in the first case,  $J_A$  according to equation (12) holds for all coefficients  $w(k)$ , in the second case  $J_C$  must be different for three different coefficient groups  $w(3k-1)$ ,  $w(3k)$  and  $w(3k+1)$ .

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A very efficient correction of the coefficients is obtained when the first and second cases are combined:

$$w_{n+1}(k) = w_n(k) - \alpha_A \cdot \text{sign}[J_A(k)] - \alpha_C \cdot \text{sign}[J_C(k)] \quad (18)$$

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As Figure 4 shows, the adaptive correction of coefficients  $w_n(k)$  according to equation (18) causes a sharp reduction in the spectral ranges outside the useful signal spectrum.

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Instead of forcing zero places in the transfer function through the correction, as shown in Figures 3 and 4, the transfer function may also be set at a constant value (e.g. 1) at certain frequencies.

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Instead of setting the transfer function of the equalizer itself at a constant value at certain frequencies, a first derivation and/or a higher derivation of the transfer function may also be set at a constant value for one or more selected frequencies.

### ABSTRACT OF THE DISCLOSURE

The coefficients of an equalizer are adapted according to an error correction algorithm so that intersymbol interference is minimized. To prevent coefficient drift in adaptive  
5 equalization, the coefficients determined with the help of the error correction algorithm are adjusted by a correction term so that the transfer function of the equalizer assumes a favorable value for one or more selected frequencies outside the useful signal frequency band.

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[10191/2232]

METHOD OF ADAPTIVE ADJUSTMENT OF THE COEFFICIENTS  
OF AN EQUALIZER

Background Information

The present invention relates to a method of adaptive adjustment of the coefficients of an equalizer, the coefficients being adapted according to an error correction algorithm so that intersymbol interference is minimized.

When digital signals are to be transmitted over channels having time-variant channel distortion, there must be adaptive equalizers in the receiver, automatically adapting to the channel distortion to compensate it. Channel distortion occurs, for example, on radio transmission channels in point-to-point radio relay connections based on multiway propagation. A distinction is made between baud-spaced equalizers and fractionally spaced equalizers. In the case of baud-spaced equalizers, there is exactly one sample at both the input and the output of each symbol clock pulse. Since the sampling condition is already violated at the input of the equalizer in this case, such equalizers have only limited capacity. Much better results are obtained with a fractionally spaced equalizer, the simplest embodiment of which processes exactly two samples per symbol clock pulse at the input. Even with a fractionally spaced equalizer, only ever one sample is calculated per symbol clock pulse at the output because there is also only ever one transmission symbol to be detected per symbol clock pulse. Since the usual error correction algorithms for adaptive adjustment of the equalizer coefficients can analyze only the baud-spaced output signal, fundamentally important information regarding the complete control of all possible coefficient settings is missing in the

case of fractionally spaced equalizers. As a result, this type of equalizer tends toward unwanted "coefficient drift" which can no longer be controlled with the usual algorithms.

Coefficient drift means that because of rounding errors in the calculation of the coefficients, there are very gradual changes in the adapted correction quantities for the coefficient values in one direction. In general, the following known algorithms are used for adaptation of the equalizer coefficients of fractionally spaced equalizers.

During the acquisition phase, as long as the transmission carrier has not yet been recognized, so the closed loop is not yet engaged for carrier phase synchronization, the constant modulus algorithm (CMA) is usually used, and the least means square algorithm (LMSA) is used during the tracking phase, i.e., in the actual continuous operation of the receiver. The two aforementioned algorithms are described, for example, in K. D. Kammeyer, Nachrichtenübertragung [Telecommunication], B. G. Teubner Verlag, Stuttgart, 1992, pp. 313-316, 510-512 and in J. G. Proakis, Digital Communications, McGraw-Hill, 1989, pp. 561-569, 587-593.

Coefficient drift is observed with the CMA algorithm in particular as a result of quantizing errors, because it is difficult especially with CMA, which is a higher order algorithm, to completely prevent offset errors as a result of quantization operations. With the LMSA algorithm, the lack of control capability, especially with dynamic transmission channels, results in inadequate adaptation of the equalizer coefficients to the channel changes. Especially channels with multiway reception, such as that with point-to-point radio relay connections, result in failure of the equalizer in cases of less serious distortion, although in terms of its basic capability, it would be quite capable of equalizing these channels.

The tap leakage algorithm (TLA), which is a variant of LMSA, is a countermeasure against the phenomenon of coefficient drift in a fractionally spaced equalizer. The tap leakage algorithm is described by R. D. Giltin, H. C. Meadors, S. B.

5 Weinstein: The Tap Leakage Algorithm: an Algorithm for the Stable Operation of a Digitally Implemented, Fractionally Spaced Adaptive Equalizer. BSDJ, no. 8, vol. 61, October 1982, pages 1817 through 1839. According to the TLA, smaller amounts are subtracted from the absolute value of the coefficients to  
10 compensate for coefficient drift. However, this measure not only prevents coefficient drift but also has a negative effect on the equalization result, i.e., there is an increase in intersymbol interference.

15 Therefore, the object of the present invention is to provide a method of the type defined in the preamble with which coefficient drift may be prevented without having a negative effect on the quality of equalization.

## 20 Advantages of the Invention

This object is achieved with the features of Claim 1 by the fact that the coefficients determined with the help of the error correction algorithm are adjusted by a correction term  
25 so that the transfer function of the equalizer assumes an invariant fixed value for one or more selected frequencies outside the useful signal frequency band. Instead of the transfer function itself, a first derivation and/or a higher derivation of the transfer function outside the useful signal  
30 frequency band may be set at a fixed value for one or more selected frequencies. This method largely reduces overshooting of the transfer function of the equalizer toward both sides of the useful signal frequency band which can be attributed to unwanted coefficient drift. This prevents dynamic fading  
35 events on the transmission link from leading to failure of the equalizer, e.g., when a certain coefficient setting can no

longer be adapted rapidly enough.

Advantageous refinements of the present invention are derived from the subclaims. The simplest possible method of calculating correction terms for the coefficients is possible if the transfer function or the first derivation and/or a higher derivation of the transfer function at the frequency  $2\pi/T$  and/or the frequency  $3\pi/T$  and/or the frequency  $4\pi/3T$  is set at a fixed value, where  $\pi/T$  is the base frequency of the useful signal frequency band. The transfer function or a first derivation and/or a higher derivation thereof may be set at the value 0 or at another fixed value at the selected frequency or frequencies.

#### Drawing

The present invention is explained in greater detail with reference to an exemplary embodiment.

Figure 1 shows a block diagram of an adaptive equalizer;

Figure 2 shows a transfer function of the equalizer without the correction according to the present invention;

Figure 3 shows a transfer function of the equalizer having a first correction, and

Figure 4 shows a transfer function of the equalizer having a second correction.

#### Description of a Embodiment

Figure 1 shows an adaptive transversal equalizer having a time-delay chain, of which first two time-delay elements V0 and V1 are shown here. A digital input signal x, which has been distorted on the transmission link from a transmitter to the receiver in which the adaptive equalizer is located, is applied to input 1 of the time-delay chain. Input signal x is delayed by  $T/2$ , T being the symbol clock pulse of input signal x, in individual time-delay elements V0, V1. Before the

individual time-delay elements, the symbols of distorted input signal  $x$  are picked up from the time-delay chain and sent to a multiplier  $M0, M1$ , where the symbol is weighted with a coefficient  $w(0), w(1)$ . All signal symbols  $y(0), y(1), \dots$   
5  $y(n)$  formed in the equalizer and weighted with coefficients  $w(0), w(1), \dots w(n)$  in this way are combined to an output signal  $y$  by a summing unit. Output signal  $y$  is sent to a decision circuit ES which decides for each symbol of output signal  $y$  which of the possible transmission symbols it most  
10 closely approximates, i.e., decision circuit ES estimates the transmission symbols most likely to be sent on the basis of the symbols of summing unit output signal  $y$ . Estimated transmission symbols  $a$  may be picked up at output 2 of decision circuit ES. Decision circuit ES also generates an  
15 error signal  $e$  which depends on the deviation between the respective symbol of summing unit output signal  $y$  and estimated transmission symbol  $a$ .

Error signal  $e$  is sent to correlators  $K0, K1$  which are  
20 responsible for forming coefficients  $w(0), w(1)$ . Correlators  $K0, K1$  determine adaptive change values for coefficients  $w(0), w(1)$  according to a known error correction algorithm, e.g., according to the LMSA algorithm mentioned above, from error signal  $e$  and the symbols of distorted input signal  $x$  picked up  
25 from the time-delay chain. Each correlator  $K0, K1$  is followed by an adder  $A0, A1$  in which a correction term  $kt(0), kt(1)$  is added to the change value output by correlator  $K0, K1$  for the coefficients  $w(0), w(1)$ . Correction terms  $kt(0), kt(1)$  are  
30 formed in a processor (PZ) according to an algorithm to be described in greater detail below. These correction terms  $kt(0), kt(1)$  are aimed at preventing the coefficient drift mentioned above.

Individual adders  $A0, A1$  are each followed by a coefficient  
35 register  $KR0, KR1$  in which integration is performed over all change values including the correction terms for coefficients

$w(0)$ ,  $w(1)$ , which then yields the respective instantaneous coefficient  $w(0)$ ,  $w(1)$ .

Figure 1 shows an adaptive equalizer for an actual digital input signal  $x$ . However, QAM signals are usually emitted in a digital radio relay system. Accordingly, four such adaptive equalizers would have to be provided for one QAM receiver, namely one in the in-phase branch, one in the quadrature-phase branch, and for compensation of crosstalk, one adaptive equalizer connected from the in-phase branch to the quadrature-phase branch and one connected from the quadrature-phase branch to the in-phase branch.

The following explains how processor PZ generates correction terms  $kt(k)$ , where  $k = 0, 1, \dots, n$ . As already indicated, coefficient drift is to be suppressed by correction terms  $kt(k)$  for coefficients  $w(k)$ . Figure 2 shows a transfer function  $E(\omega)$  of an equalizer, where the useful signal frequency band has its base frequencies at  $\omega = \pm\pi/T$ , but the equalizer may influence the spectrum in the range of  $\omega = -2\pi/T$  to  $\omega = +2\pi/T$ . Unwanted coefficient drift is manifested by a gain in the spectral ranges not used by the useful signal (shown in gray in Figure 2). Correction term  $kt(k)$  directly influences the transfer function of the equalizer, so that the overshooting outside the useful signal frequency band is reduced and therefore no more coefficient drift occurs. At the same time, however, the transfer function within the useful signal frequency band remains completely unaffected by this, so that the actual equalization is not impaired.

The following equation holds for transfer function  $E(\omega)$  of the equalizer:

$$\begin{aligned}
E(\omega) &= \sum \left( w_i(k) + jw_q(k) \right) e^{-j\omega k T/2} \\
&= E_i(\omega) + jE_q(\omega) \\
&= \sum w_i(k) \cos \omega k \frac{T}{2} + \sum w_q(k) \sin \omega k \frac{T}{2} \\
&\quad + j \left( -\sum w_i(k) \sin \omega k \frac{T}{2} + \sum w_q(k) \cos \omega k \frac{T}{2} \right)
\end{aligned} \tag{1}$$

where  $w_i(k)$  and  $w_q(k)$  are the  $k$ -th in-phase coefficient and quadrature-phase coefficient, respectively.

Summation  $\Sigma$  extends over all coefficients from  $k = 0$  through  $k = n$ . The algorithm being run in processor PZ should form a zero place or another fixed value of the transfer function at at least one specific frequency  $\omega_0 \approx \pm\pi/T$  to reduce the transfer function outside the useful frequency band. This yields as the target function of the algorithm:

$$|E(\omega_0)| = |E_i(\omega_0)|^2 + |E_q(\omega_0)|^2 \tag{2}$$

With a complex equalizer for QAM signals, all the respective four partial equalizers are to be adjusted in their transfer function independently of one another in the sense described above, so the differentiation between  $w_i$  and  $w_q$  for the in-phase branch and the quadrature-phase branch may be omitted. This yields the following for the absolute value of the transfer function:

$$|E(\omega_0)|^2 = \left| \sum w(k) \cos \omega_0 k \frac{T}{2} \right|^2 + \left| \sum w(k) \sin \omega_0 k \frac{T}{2} \right|^2 \tag{3}$$

and for its gradient:

$$\begin{aligned} \frac{\partial}{\partial w(k)} |E(\omega_0)|^2 &= 2|E_i(\omega_0)| \frac{\partial |E_i(\omega_0)|}{\partial w(k)} + 2|E_q(\omega_0)| \frac{\partial |E_q(\omega_0)|}{\partial w(k)} \\ &= 2\cos\omega_0 k \frac{T}{2} \sum w(k) \cos\omega_0 k \frac{T}{2} + 2\sin\omega_0 k \frac{T}{2} \sum w(k) \sin\omega_0 k \frac{T}{2} \end{aligned} \quad (4)$$

The algorithm for correction of coefficients  $w(k)$  is then formed as follows:

$$w_{n+1}(k) = w_n(k) - \alpha \cdot \text{sign}[J(k)] \quad (5)$$

$w_n(k)$  is the coefficient formed one clock pulse previously, and  $w_{n+1}(k)$  is the instantaneous coefficient derived by addition of correction term  $kt(k) = -a \cdot \text{sign}[J(k)]$  to coefficient  $w_n(k)$ . Otherwise, the change value for coefficient  $w_n(k)$  formed according to the known CMA or LMSA algorithm, for example, in correlators K0, K1 is not taken into account in equation (5) for the sake of simplicity.

In equation (5):

$$J(k) = \frac{\partial}{\partial w(k)} |E(\omega_0)|^2 \quad (6)$$

$$J(k) = W_C \cdot \cos\omega_0 k \frac{T}{2} + W_S \cdot \sin\omega_0 k \frac{T}{2}$$

where the abbreviations

$$\begin{aligned} W_C &= \sum w(k) \cos\omega_0 k \frac{T}{2} \\ W_S &= \sum w(k) \sin\omega_0 k \frac{T}{2} \end{aligned} \quad (7)$$

were used. Since only a slight influence on the coefficients is desired (small  $\alpha$ ), the algorithm may be simplified in practical operation to a signum form. Effectiveness factor  $\alpha$  for correction term  $kt(k)$  is adjusted to a suitable value by field simulation.

The algorithm is to be implementable without any great effort. Therefore, it must be limited to frequencies  $\omega_0$  which permit a simple and most periodic possible calculation of the trigonometric functions. The following frequencies may be used for this:

$$\omega_A = \frac{2\pi}{T}, \omega_B = \frac{3\pi}{2T} \text{ and } \omega_C = \frac{4\pi}{3T}. \quad (8)$$

Two simplest cases  $\omega_A$  and  $\omega_C$  shall now be discussed.

1<sup>st</sup> case:  $\omega_A = 2\pi/T$

For this especially simple case, it holds that

$$\begin{aligned} \sin \omega_A k \frac{T}{2} &= \sin k\pi = 0 \\ \cos \omega_A k \frac{T}{2} &= \cos k\pi = (-1)^k \end{aligned} \quad (10)$$

and therefore

$$\frac{\partial}{\partial w(k)} |E(\omega_A)|^2 = 2(-1)^k \cdot \sum w(k)(-1)^k \quad (10)$$

The complete algorithm for adaptation of the coefficients is thus written as follows according to equation (5):

$$w_{n+1}(k) = w_n(k) - 2\alpha \cdot \text{sign}\left[(-1)^k \sum w(k)(-1)^k\right] \quad (11)$$

where

$$J_A = 2(-1)^k \cdot \sum w_n(k)(-1)^k \quad (12)$$

Since the middle coefficients are preferably involved in the unwanted components of the transfer function, the algorithm according to equation (11) may be limited to an interval on the order of  $k \in [-4, 4]$ . All coefficients must of course be taken into account in equation (12).

Figure 3 shows the result of this algorithm, where it is apparent that there has already been a certain reduction in the transfer function outside the useful signal frequency band in comparison with the uncorrected transfer function shown in Figure 1.

$$\text{2nd case: } \omega_c = \frac{4\pi}{3T}$$

For this case, it holds that

$$\begin{aligned} \sin \omega_c k \frac{T}{2} &= \sin \frac{2}{3} k \pi = \Im(z^k) \\ \cos \omega_c k \frac{T}{2} &= \cos \frac{2}{3} k \pi = \Re(z^k) \end{aligned} \quad (13)$$

$$z = e^{j2\pi/3} = e^{j120^\circ}$$

With the table:

K	-4	-3	-2	-1	0	1	2	3	4
$\sin \frac{2}{3}k\pi$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$	$-\sqrt{0.75}$	0	$\sqrt{0.75}$
$\cos \frac{2}{3}k\pi$	-0,5	1	-0,5	-0,5	1	-0,5	-0,5	1	-0,5

5

it holds that:

$$W_C = \sum w(k) \cos \omega_c k \frac{T}{2} = \sum w(3k) - 0,5 \sum [w(3k+1) + w(3k-1)] \quad (14)$$

$$W_S = \sum w(k) \sin \omega_c k \frac{T}{2} = \sqrt{0,75} \sum [w(3k+1) - w(3k-1)] \quad (15)$$

To calculate the correction terms, additional abbreviations are introduced:

$$\begin{aligned} W_{-1} &= \sum w(3k-1) \\ W_0 &= \sum w(3k) \\ W_{+1} &= \sum w(3k+1) \\ WD &= W_{+1} - W_{-1} \\ WS &= W_{+1} + W_{-1} \end{aligned} \quad (16)$$

It thus holds that:

$$\begin{aligned}
J_C(3k-1) &= -\sqrt{0,75} \cdot W_s - 0.5 \cdot W_C = \frac{1}{4}(-3WD - 2W_0 + WS) \\
J_C(3k) &= W_C \\
J_C(3k+1) &= +\sqrt{0,75} \cdot W_s - 0.5 \cdot W_C = \frac{1}{4}(+3WD - 2W_0 + WS)
\end{aligned}
\tag{17}$$

Whereas in the first case,  $J_A$  according to equation (12) holds for all coefficients  $w(k)$ , in the second case  $J_C$  must be different for three different coefficient groups  $w(3k-1)$ ,  $w(3k)$  and  $w(3k+1)$ .

A very efficient correction of the coefficients is obtained when the first and second cases are combined:

$$w_{n+1}(k) = w_n(k) - \alpha_A \cdot \text{sign}[J_A(k)] - \alpha_C \cdot \text{sign}[J_C(k)] \tag{18}$$

As Figure 4 shows, the adaptive correction of coefficients  $w_n(k)$  according to equation (18) causes a sharp reduction in the spectral ranges outside the useful signal spectrum.

Instead of forcing zero places in the transfer function through the correction, as shown in Figures 3 and 4, the transfer function may also be set at a constant value (e.g. 1) at certain frequencies.

Instead of setting the transfer function of the equalizer itself at a constant value at certain frequencies, a first derivation and/or a higher derivation of the transfer function may also be set at a constant value for one or more selected frequencies.

What is claimed is:

1. A method of adaptive adjustment of the coefficients of an equalizer, the coefficients ( $w(0)$ ,  $w(1)$ ) being defined according to an error correction algorithm so that intersymbol interference is minimized, wherein the coefficients ( $w(0)$ ,  $w(1)$ ) determined with the help of the error correction algorithm are adjusted by a correction term ( $k(0)$ ,  $k(1)$ ) so that the transfer function ( $E(\omega)$ ) of the equalizer or a first derivation and/or a higher derivation of the transfer function assumes an invariant fixed value for one or more selected frequencies outside the useful signal frequency band.

2. The method according to Claim 1, wherein the coefficients ( $w(0)$ ,  $w(1)$ ) are adjusted so that the transfer function ( $E(\omega)$ ) or a first derivation and/or a higher derivation of the transfer function assumes a fixed value at the frequency  $2 \pi/T$  and/or at the frequency  $3 \pi/T$  and/or at the frequency  $4 \pi/3T$ ,  $\pi/T$  being the limit frequency of the useful signal frequency band.

3. The method according to one of Claims 1 or 2, wherein the transfer function ( $E(\omega)$ ) or a first derivation and/or a higher derivation of the transfer function is/are set at the value 0 or at another constant value at the selected frequency or frequencies.

## Abstract

The coefficients  $(w(0), w(1))$  of an equalizer are adapted according to an error correction algorithm so that intersymbol interference is minimized. To prevent coefficient drift in adaptive equalization, the coefficients  $(w(0), w(1))$  determined with the help of the error correction algorithm are adjusted by a correction term  $(k(0), k(1))$  so that the transfer function of the equalizer assumes a favorable value for one or more selected frequencies outside the useful signal frequency band.

(Figure 1)

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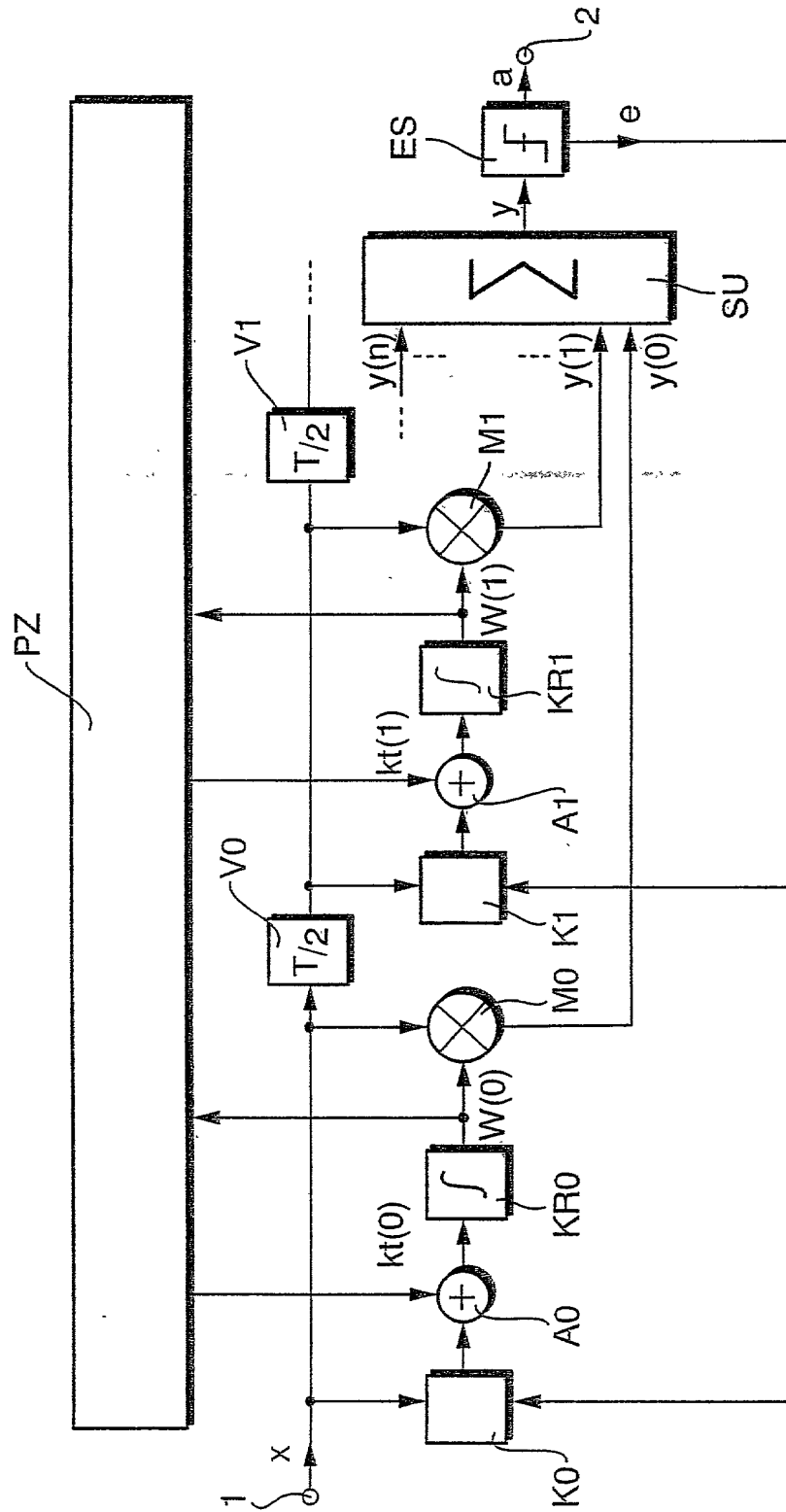


Fig. 1

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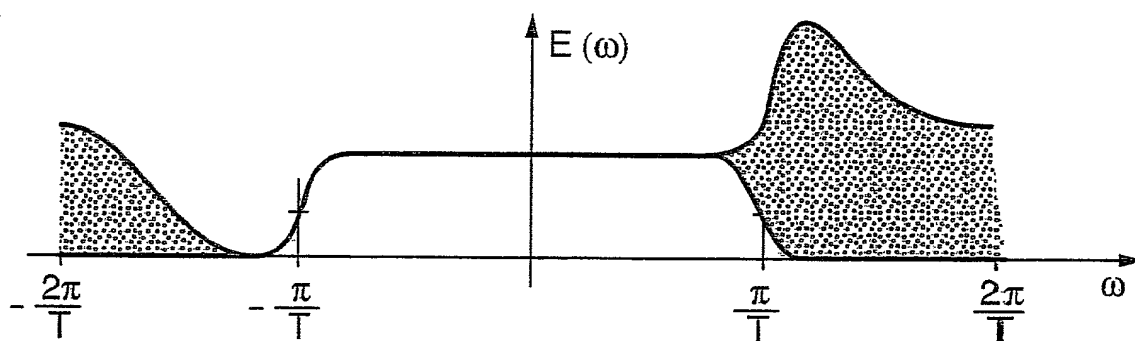


Fig. 2

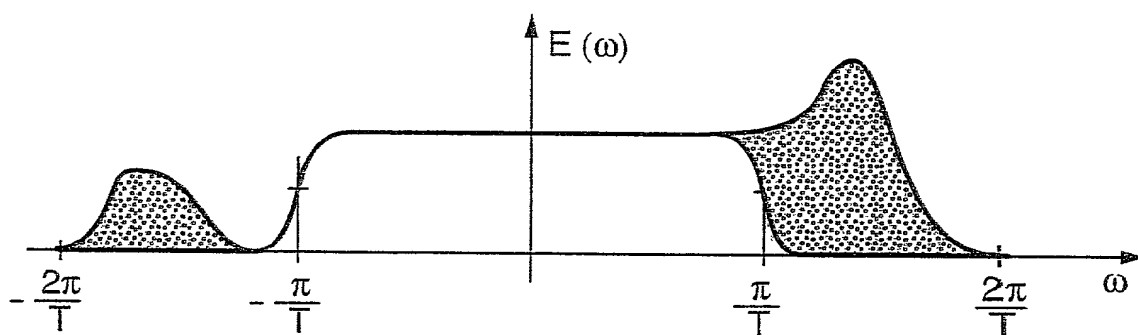


Fig. 3

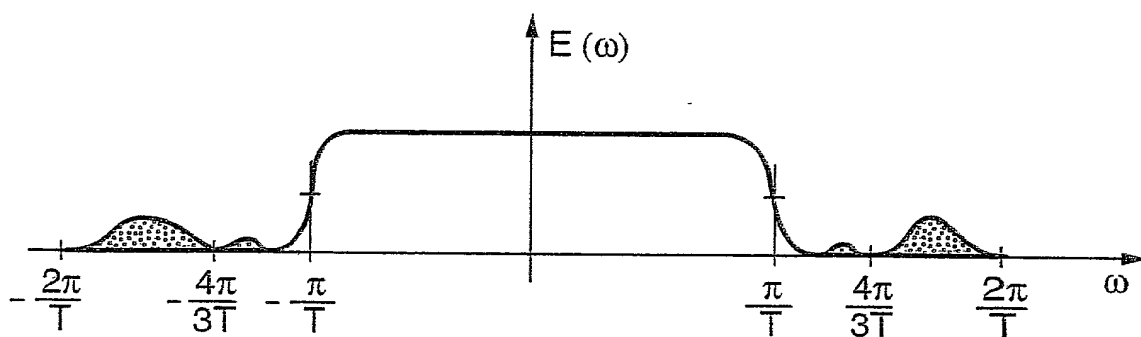


Fig. 4

DECLARATION AND POWER OF ATTORNEY

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name.

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled **METHOD OF ADAPTIVE ADJUSTMENT OF THE COEFFICIENTS OF AN EQUALIZER**, the specification of which was filed as International Application **PCT/DE00/02411** on July 20, 2000;

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claims.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, § 1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, § 119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application(s) for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

2L244510569

PRIOR FOREIGN APPLICATION(S)

Number	Country filed	Day/month/year	Priority Claimed Under 35 USC 119
DE 19934672.0	Fed. Rep. of Germany	23 July 1999	Yes

2 And I hereby appoint Richard L. Mayer (Reg. No. 22,490) and Gerard A. Messina (Reg. No. 35,952) my attorneys with full power of substitution and revocation, to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith.

Please address all communications regarding this application to:

KENYON & KENYON  
One Broadway  
New York, New York 10004  
**CUSTOMER NO. 26646**

Please direct all telephone calls to Richard L. Mayer at (212) 425-7200.

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful and false statements may jeopardize the validity of the application or any patent issued thereon.

1-00  
Inventor: Gerhard HERBIG

Inventor's Signature: Gerhard P. Herbig

Date: 22nd of February 2002

Residence: Reutenhofst. 42/2  
71570 Oppenweiler DEX  
Federal Republic of Germany

Citizenship: Federal Republic of Germany

Post Office Address: Same as above.

2:00  
Inventor: Thomas GEBAUER

Inventor's Signature: Thomas Gebauer

Date: 2/22/2022

Residence: ~~Im Reutle 36~~ Bergäckerstr. 6  
71573 Allmersbach i. Tal  
Federal Republic of Germany

DEX

Citizenship: Federal Republic of Germany

Post Office Address: Same as above.

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